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(54) Dynamic equalizing/filtering device

(57) Equalizing/filtering device associated with a reproduction and/or amplifying system which dynamically changes its frequency response in the time domain in connection with the amplitude variation of the input or output signal from the said device or the sound pressure listening level to the aim of taking into account the nonlinearity of the response of the human ear at the variation of said parameters.

The frequency response of said equalizer/filter if therefore calculated according to the audiometric diagram curves normal to the human's ear (figure 1) which reproduce the frequency response to the ear itself.

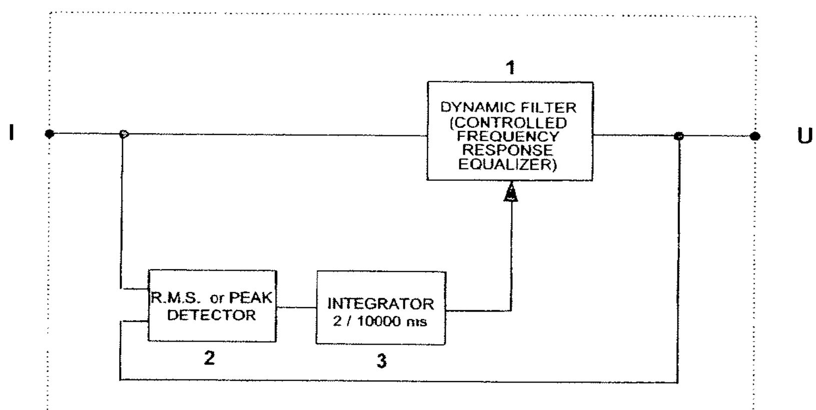

FIG. 2

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Description

The present invention refers to a dynamic equalizing/filtering device for achieving the right linearity correction of audio reproduction and/or amplifying systems.

Up to date the audio systems used for audio signal recording use a direct proportional linearity between the audio signal to be recorded and the signal reported onto the recording layer. The recorded audio signals thus reproduced will provide a different sound sensation to the human ears than the one that should be heard during the listening to the direct sound source.

Actually, with reference to figure 1, the normal audiogram is shown and it can be seen how the frequency response of the human ear varies with the loudness level (having a different loudness perception on the audible audio spectrum), and the graph shows how the ear does not have the same loudness perception with the same intensity on all the audible frequency spectrum.

More precisely, at low loudness levels, the low (20-200Hz) and high (5-20 Khz) frequencies are penalized by several decibels in respect to the medium frequencies, while at higher loudness levels the ear frequency response gets always more linear.

At present the devices available on the market intervene on the reproduction and/or amplification frequency response system to increase the reproduction quality modifying in a static manner the input signal.

As an example, the device known as loudness control (physiologic loudness control on the amplifiers) filters the input signal to be reproduced increasing the low and high frequencies with respect to the medium frequencies. Generally this enhancement is reduced as the audio reproduction and/or amplification system volume level increases and it is definitively excluded at a certain level of the volume itself. Thus, once the audio reproduction and/or amplification system volume level has been fixed, the filtering action of the signal will remain constant for any variation of the signal to be reproduced.

Other known devices as tone controls, graphic or parametric equalizers, provide a constant correction of the audio reproduction system frequency response in a static way independently from any input or output signal variation or from the volume control providing a correction of any possible non-linearity in the audio amplifying system or with respect to the listening room acoustic absorption or reflections.

Instead other known devices as digital ambience equalizers (D.S.P. Digital Signal Processor) are used to recreate or to modify the ambience response where the audio reproduction takes effect, adding or deleting echoes, reverberations, etc, intervening in the time and/or frequency domain always in a static manner.

Object of the invention is to obtain an audio signal reproduction the more accurate possible with respect to the effective audio perception of the human ear, neither depending upon the recording conditions in which the signal was recorded, nor depending on the amplifying/reproducing instruments used for recording purposes.

This object is obtained by means of the dynamic filtering/equalizing device, according to the invention, whose features are listed in the attached claim 1. Advantageous embodiments of the invention are explained in the dependent claims.

Substantially, the device according to the invention corrects the linearity of any amplifying and/or reproducing system (high fidelity or not, monophonic single channel, stereophonic or polyphonic etc. analog and/or digital exclusively dedicated to sound amplification or used in combination with other systems such as video systems, data systems and so on) producing, as a final result, a time domain variable system frequency response as a function of (a) the input signal amplitude and/or (b) the output signal amplitude and/or (c) the effective acoustic sound pressure level at the listener's ear in order to consider the non-linearity of the human ear frequency response as a function of the previous mentioned variables.

Further features of the invention will be clarified by the following detailed description with reference to an exemplary, non limitative embodiment of the invention, shown in the accompanying drawings, wherein:

figure 1 is the normal loudness audiometric diagram that shows the sound pressure levels of the human ear as function of the audio frequencies;

figure 2 is a block diagram of the equalizing/filtering device according to the invention;

figure 3 is a block diagram of the equalizing/filtering device according to the invention realized in a digital form;

figures 4 and 5 are two block diagrams of the equalizing/filtering device according to the invention realized in similar form;

figure 6 shows an amplifying system chain according to the prior art.

Figure 1 shows that once a sound pressure level and frequency is determined, it is assigned the relevant curve that is the real human ear frequency response for those conditions.

To obtain the more accurate possible reproduction and/or amplification frequency response as function of the human ear perception, the amplifying system will have to respond for low loudness levels as close as possible to the relevant curve determined on the loudness audiometric diagram shown in figure 1, while it will have to dynamically adapt to an always more linear frequency

response as the sound pressure level increases.

The device according to the present invention, whose block diagram is shown in figure 2, will set up the dynamic filter 1 frequency response as a function of the input signal I that can either be the sound pressure level or the input or output signal amplitude of the amplifying and/or reproduction system, for the best result based on the optimal curve chosen from the audiometric diagram shown in figure 1. The audiometric curve choice is based upon the frequency band of the input signal I on to which the device will have to intervene, thus measuring the input I and/or output U level signal in the relevant band, as an example by means of an R.M.S. (Root-Mean-Square) or peak detector 2. The value obtained is kept constant by the next integrator 3, for the set period of time for the device according to the invention, to achieve the correction on the signal I. A variation of the signal I will change the value measured by the detector 2 providing a change of the optimal curve of the audiometric diagram shown in figure 1, thus the filter 1 will set its frequency response related to the new curve in the way that the overall frequency response of the full amplifying/reproducing system will dynamically adapt to the input signal I as to obtain the best reproduction of the signal itself.

An embodiment of the dynamic equalizing/filtering device implemented through a digital technology according to the present invention, where only one channel is shown is reported on figure 3. The analog input signal I1, filtered into the interested band, is sampled and converted by an analog/digital (A/D) converter 7 in digital signals and sequentially stored by a D.S.P. (Digital Signal Processor) 4 in a dynamic memory, as an example RAM (Random Access Memory) 5.

The storable number of samples depends upon the maximum RAM 5 capacity and gives the effective quality of the final result.

The D.S.P. 4 by means of working program codes stored, as an example, into a EPROM (Erasable Programmable Read Only Memory) 6, is able to: a) verify a peak or mean variation of the samples (input signal increasing or decreasing trend) and b) decide which is the peak or mean input level at that moment.

Into the EPROM 6, further to the D.S.P. 4 working program codes, are stored also codes (CODE P/F) concerning the relation between the sound pressure levels and the audible frequency range of the human ear perception reported into the audiometric diagram shown in figure 1. Practically for each frequency response curve it is assigned a CODE P/F.

Particularly a (CODE P/F) code table of a hundred of values can almost cover the 120 dB dynamic pressure levels range starting from the minimum audible threshold level and the maximum pain threshold level.

Once the mean or peak input level is found to obtain the absolute output pressure level value produced at the listener position, the D.S.P. 4 can be programmed, by means of a keyboard 11, to interpolate the mean or peak input level as a function (a) of the mean efficiency value of the loudspeaker system used (data normally supplied by the manufacturer), (b) of the mean distance from the loudspeaker system to the listener position, or even (c) to the amplifying system voltage gain factor.

Through this interpolation the D.S.P. 4 computes the absolute loudspeaker output pressure level value as function of the input signal, choosing the functional intermediate point A with relation to one of the curve shown in figure 1. The absolute level is then compared with the code table (CODE P/F) stored into the EPROM 6 to choose the correct frequency response curve to apply in order to modify the input signal.

The D.S.P. 4, using the CODE P/F chosen, will at this point initialize the frequency response of its digital filter to modify the output signal frequency response. The D.S.P. 4 will then supply to the digital filter the digital sampled signal previously stored in RAM 5 using the FIFO (first in, first out) modality. At the digital filter output the content of the samples of the digital signal will be modified in respect to the input as per the present filter setting. The modified digital samples will be sent to the digital to analog (D/A) 8 converter that will convert the digital signal back to the analog form.

Furthermore the D.S.P. 4 function will be to continuously modify the digital filter action (internally or externally implemented) setting continuously different frequency responses variable in the time frame as a function of the processed samples and of the mean or peak audio pressure level of said samples.

A more accurate realization of the device according to the present invention, can provide the measurement of a feedback signal 13, through the attenuator 14, coming from the loudspeaker system or from other devices of the amplifying system chain or through an external connection of a microphone 9, placed nearby the listener, that picks-up the acoustic signal produced by the reproduction and/or amplifying system and supply it to the A/D converter 10. In this way the D.S.P. 4 computing is directly made on the real acoustic output pressure level instead than on the computed evaluated level.

The above mentioned A/D 7, D/A 8 and D/A 10 converters can be eventually internally implemented into the D.S.P. 4.

Moreover through the keyboard 11 it is possible to set other parameters that interact with the D.S.P. 4 to modify: a) the filter response as a function of the listener age and human race as the human ear frequency response is also a function of the above mentioned parameters, b) the filter time response as a function of the listener pleasure or room time response, c) the loudspeaker system parameters. The post-conversion filter 12 is needed for anti-aliasing and spurious harmonic suppression over 20 Khz produced by the D/A 8 converter during its normal working.

Figure 4 refers to a block diagram of an analog realization of the device where only one audio channel is shown and the functionality is obtained by means of fre-

quency band division, where the input signal I2 is filtered by a series of band-pass filters PB1,......, PBn, low pass filter PB and high pass filter PA which are amplitude and/or frequency controlled.

The input signal I2 is also filtered by the filters block consisting of the low pass filter B, high pass filter A and band pass filters B1,....., Bn tuned in the same band of the above mentioned PB, PA, PB1..., PBn filters, then passed through peak or R.M.S. detectors R1,....., Rn+2, to evaluate the signal amplitude for each filter in the selected band. The signal is then passed through integrators S1,...., Sn+2 to maintain at a constant level the detector signals for the time frame needed for the filter to function correctly. The values obtained are used to set the next components with logarithmic characteristics L1,..., Ln+2 that have the function to increase or decrease the signal level processed by the filters PB1,...,PBn, PB and PA to obtain the correct response on the relevant bands following the audiometric diagrams or part of the same as shown in figure 1.

The output signals coming from the PB1,...,PBn, PB and PA filters is then recomposed by an output analog summing amplifier.

Figure 5 shows a block diagram of an analog realization of the device, where only one audio channel is shown where the functionality is obtained intervening directly on the filters of a loudspeaker system where the input signal I3 is filtered by a series of band pass filters FPB1,..., FPBn, low pass filter FPB and high pass filter FPA whose amplitude and/or frequency response is opportunely corrected by the dynamic active or passive regulators RD1,...,RDn+2, whose function is to increase or decrease the signal level that feeds the loudspeakers or drivers to obtain the correct response on the interested bands following the audiometric diagrams or part of the same as shown in figure 1.

Figure 6 shows a chain of components normally used for reproduction and/or amplification purposes of an audio signal. The present invention covers the use of any technical realization inserted in any point of the reproduction and/or amplification system whose purpose is to dynamically regulate the frequency response of the entire system or of any component of the same.

The device according to the present invention can be applied to any stereophonic or polyphonic amplifying system and can be inserted in any position signed, from point 1 to X+m of the amplifying system chain shown in figure 6 as a sole component or as integral part of any of the components signed from A to N+1.

A more limited realization of the present invention can clearly be realized without operating on all the audible human frequency range, but working only on a more limited portion of the audible spectrum frequency range.

Claims

1. An equalizing/filtering device associated to a reproduction/amplifying audio system that elaborates the frequency response, characterized in that said device dynamically changes continuously the frequency response in relation to the input or output signal amplitude of the said audio system or in relation to the listening sound pressure level.

2. An equalizing/filtering device according to claim 1 applied to take in account the non-linearity of the human ear frequency response.

3. An equalizing/filtering device according to claims 1 or 2 characterized in that the said frequency response is elaborated in a precise or approximated manner according to the curves of the loudness audiometric diagram of the human ear (figure 1), based upon the interested band and on the amplitude of the input signal in that band.

4. An equalizing/filtering device according to claim 3, characterized in that it works on a partial band of the audible frequency spectrum of the said audiometric diagram (figure 1).

5. An equalizing/filtering device according to claims 3 or 4, characterized in that it works on a partial band of sound pressure levels of said audiometric diagram (figure 1).

6. A device according to anyone of claims 3 to 5, characterized in that it comprises means for varying the period of time established for the intervention on the signal to be elaborated.

7. A device according to anyone of claims 3 to 6, characterized in that it comprises means to vary the curves of the said audiometric diagram.

8. A device according to any of the previous claims, characterized in that it is implemented by a D.S.P. (Digital Signal Processor) (4) that elaborates signals at its input according to the audiometric curves stored into a memory, particularly a EPROM (6).

9. A device according to claim 8, characterized in that it comprises analog to digital converters (7, 10) on the input stage and digital to analog converters (8) on the output stage.

10. A device according to claim 8, characterized in that the signal elaborated by analog to digital converters (7, 10) is stored into a dynamic memory, particularly a RAM (5).

11. A device according to any of claims 1 to 7, characterized in that it is implemented with a number of filters (PB1,....,PBn,PB,PA) and (B1,....,Bn,B,A) that analyse the audible frequencies with detectors (R1,....,Rn+2) that evaluate the signal amplitude on

any frequency analysed with integrators (S1,...,Sn+2) that momentarily store this value that is used to tune the next components particularly with logarithmic characteristics (L1,...,Ln+2).

12. A device according to any of claims 1 to 7, characterized in that it is implemented directly on filters (FPB1,...,FPBn,FPB,FPA) of a loudspeaker system by means of dynamic regulator devices (RD1,...,RDn+2).

13. A device according to claim 12, characterized in that the dynamic regulator devices (RD1,...,RDn+2) are active or passive, and eventually comprised into the filters (FPB1,...,FPBn, FPB, FPA) or in some of the same.

14. A device according to claims 12 and 13, characterized in that the dynamic regulator devices (RD1,...,RDn+2) intervene in a complete or partial manner varying the amplitude of the frequency as function of the input signal.

15. A device according to anyone of the previous claims, characterized in that the elaborated signals are controlled by an external microphone (9) and/or by a feedback signal coming from any of the components of the amplifying and/or reproduction system chain.

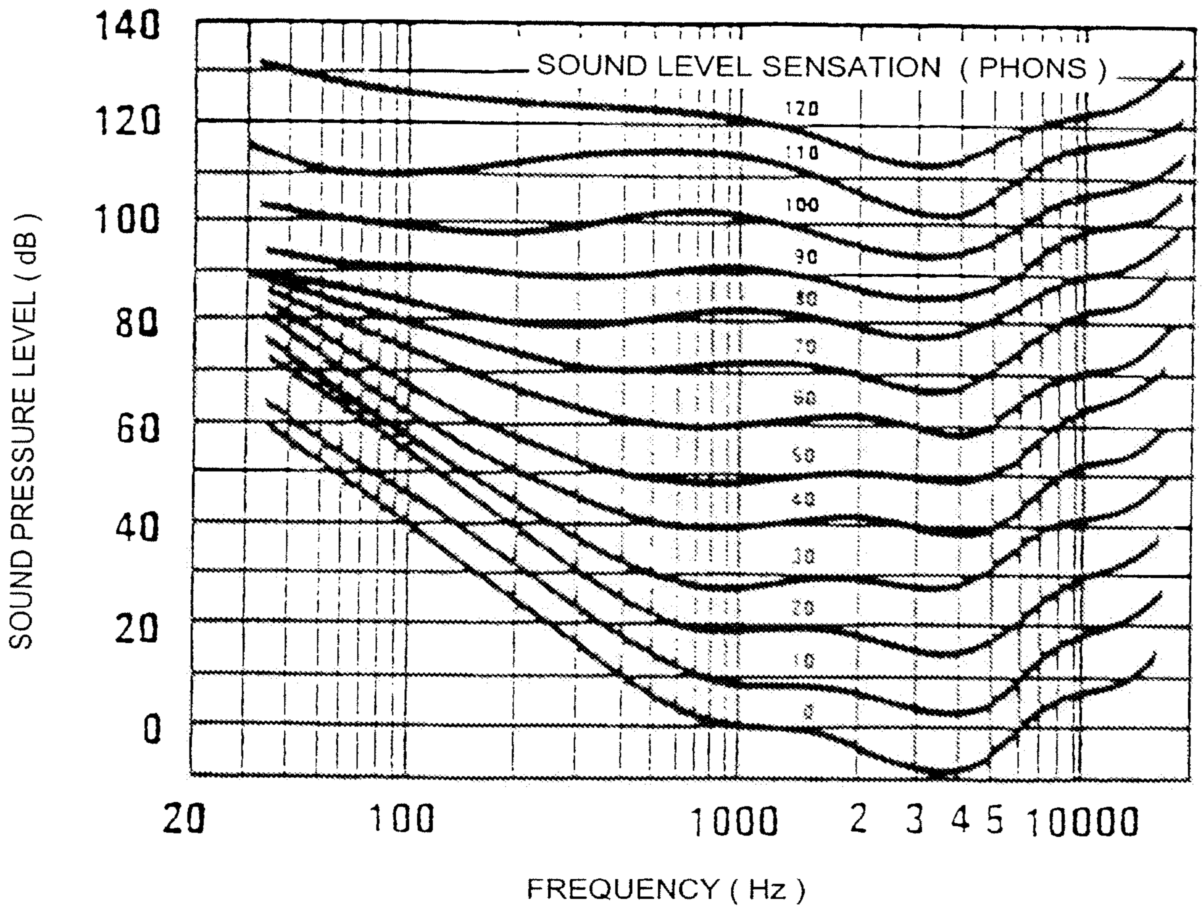

140
120
100
80
60
40
20
0
SOUND PRESSURE LEVEL (dB)
SOUND LEVEL SENSATION (PHONS)
120
110
100
90
80
70
60
50
40
30
20
10
0
20
100
1000
2
3
4
5
10000
FREQUENCY (Hz)

FIG. 1

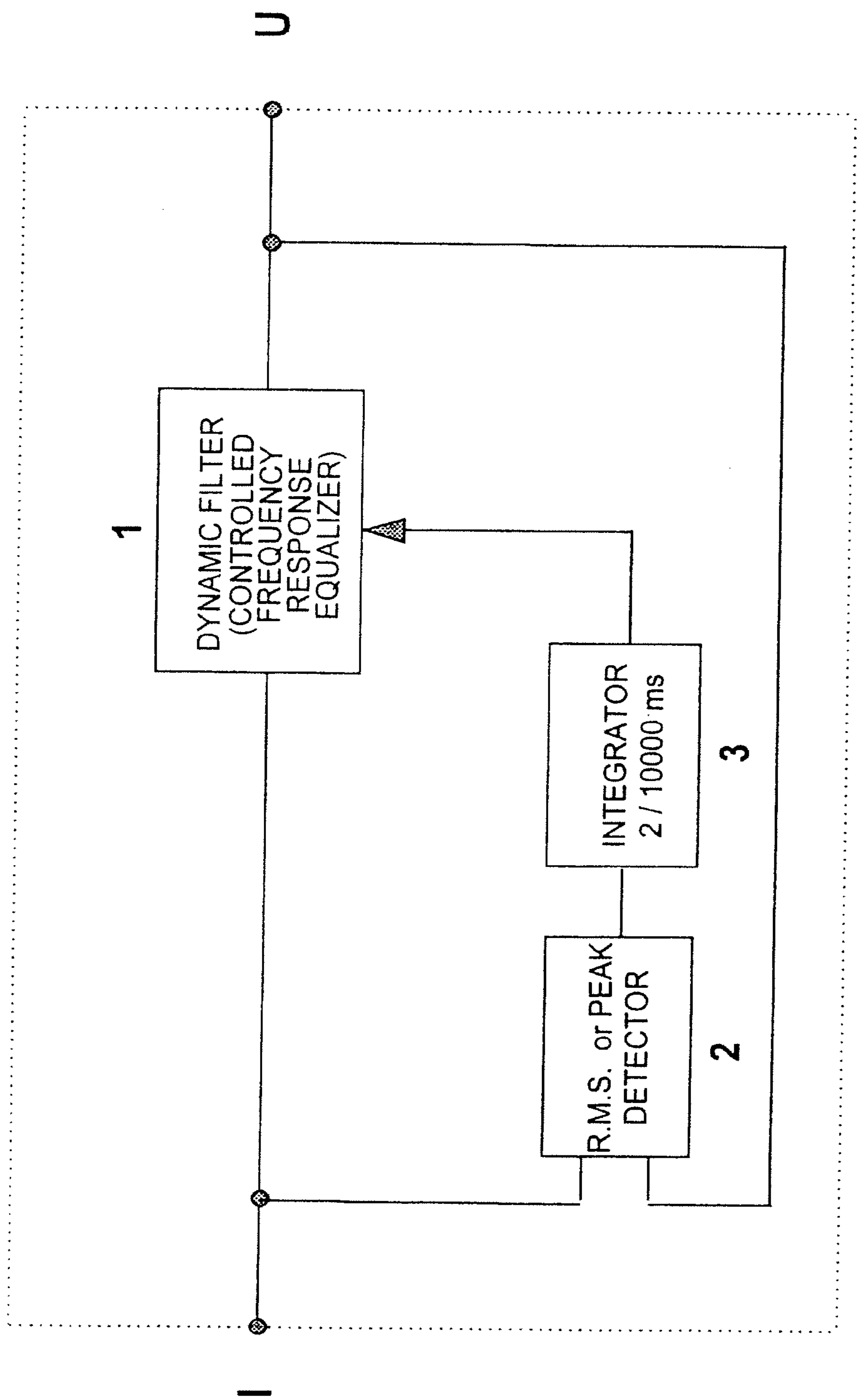

U
1
DYNAMIC FILTER
(CONTROLLED
FREQUENCY
RESPONSE
EQUALIZER)
INTEGRATOR
2 / 10000 ms
3
R.M.S. or PEAK
DETECTOR
2
I

FIG. 2

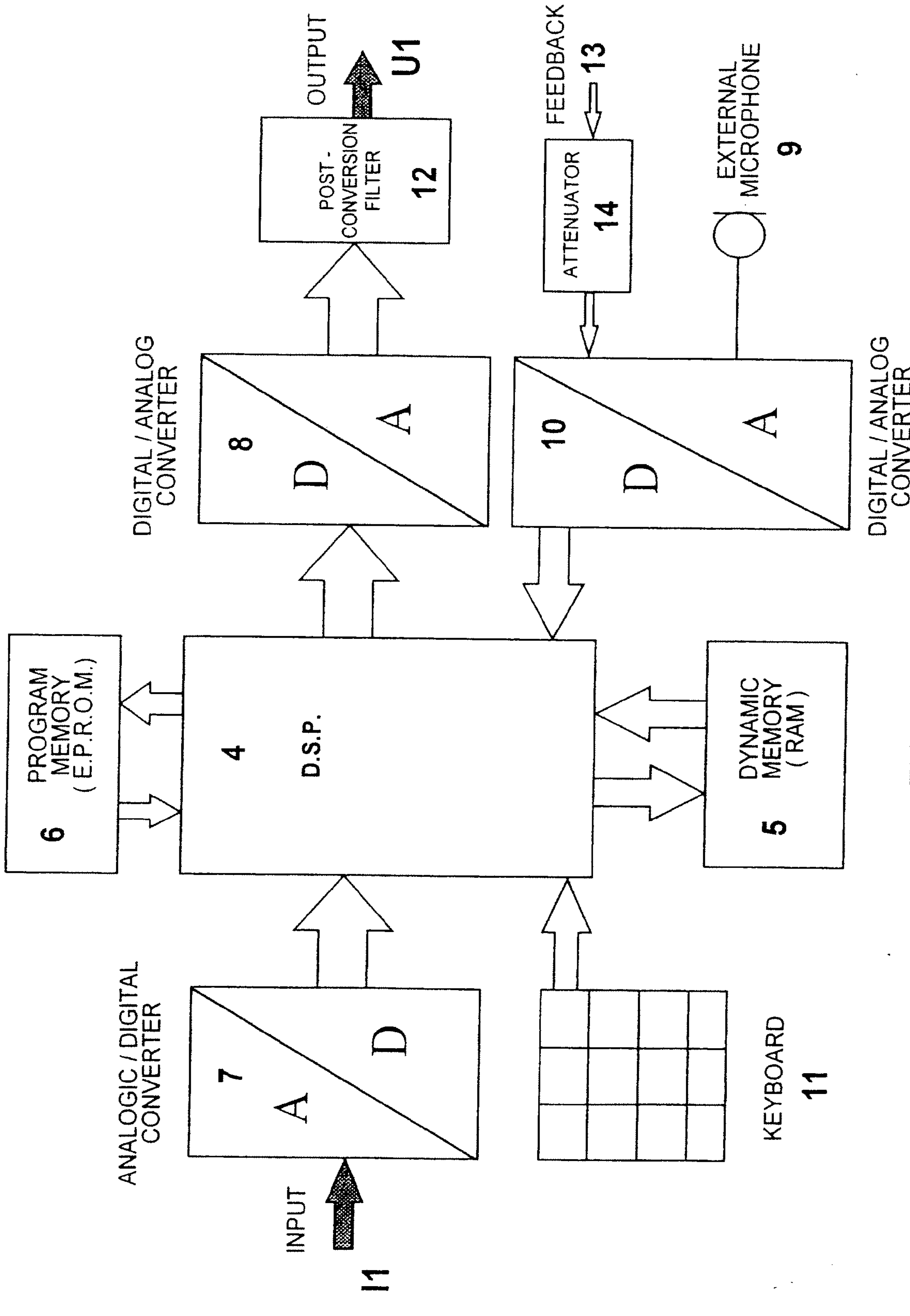

INPUT
I1
ANALOGIC / DIGITAL
CONVERTER
7
A
D
KEYBOARD
11
PROGRAM
MEMORY
(E.P.R.O.M.)
6
4
D.S.P.
DYNAMIC
MEMORY
(RAM)
5
DIGITAL / ANALOG
CONVERTER
8
D
A
10
D
A
DIGITAL / ANALOG
CONVERTER
POST -
CONVERSION
FILTER
12
OUTPUT
U1
FEEDBACK
13
ATTENUATOR
14
EXTERNAL
MICROPHONE
9

FIG. 3

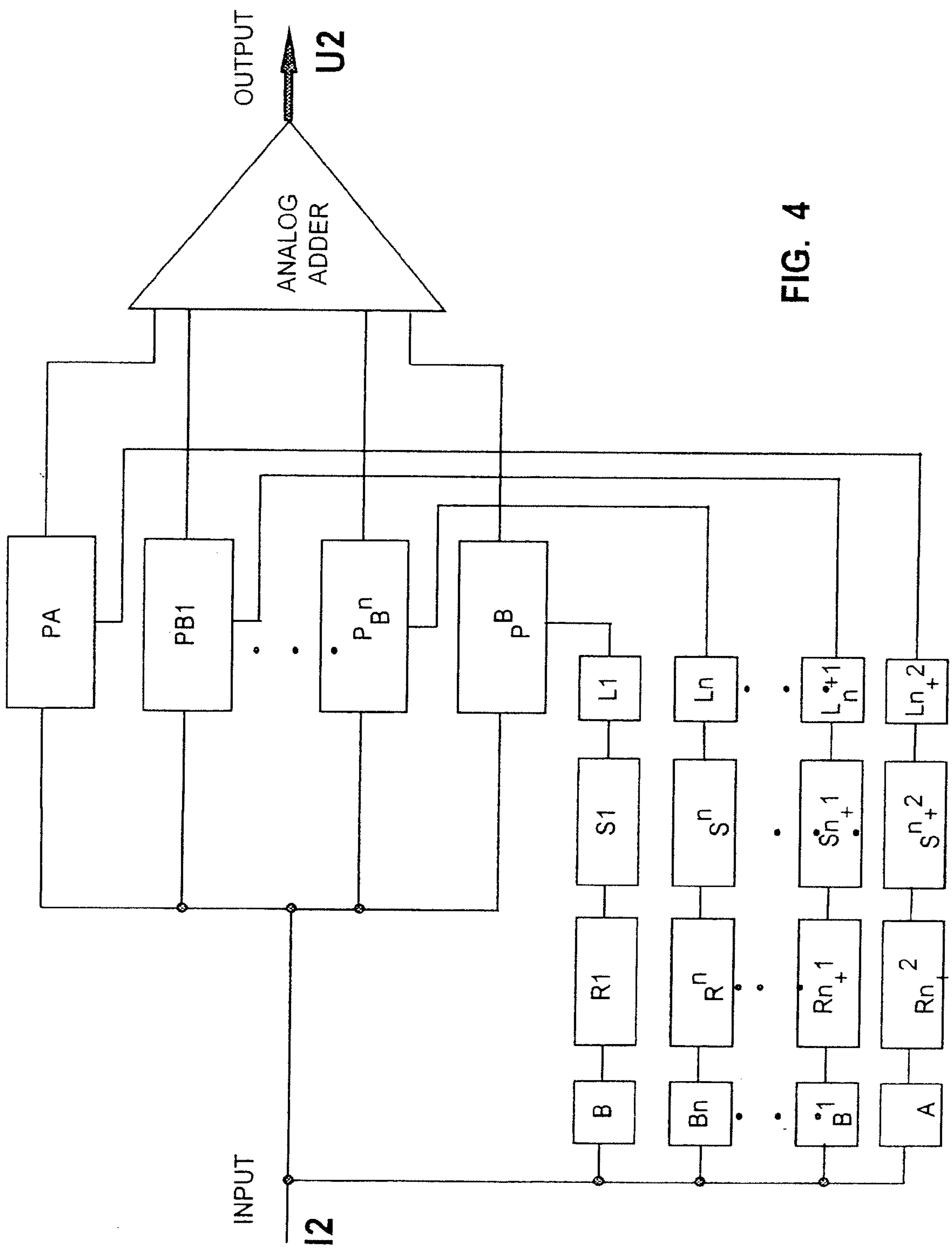

OUTPUT
U2
ANALOG
ADDER
PA
PB1
P_B^n
P^B
L1
Ln
L_{n+1}
Ln_{+2}
S1
S^n
Sn_{+1}
S^n_{+2}
R1
R^n
Rn_{+1}
Rn_{+2}
B
Bn
B^1
A
INPUT
I2

FIG. 4

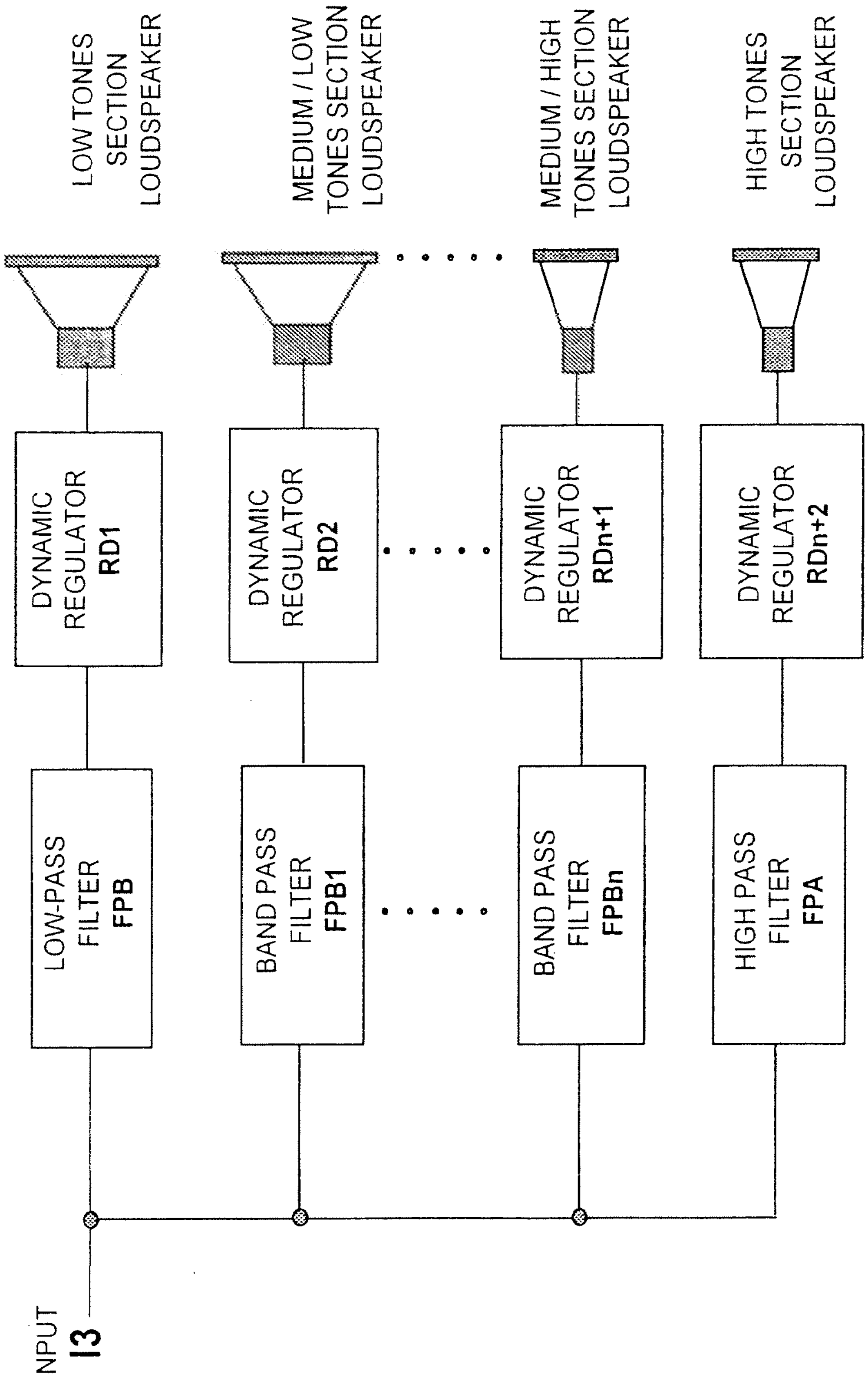
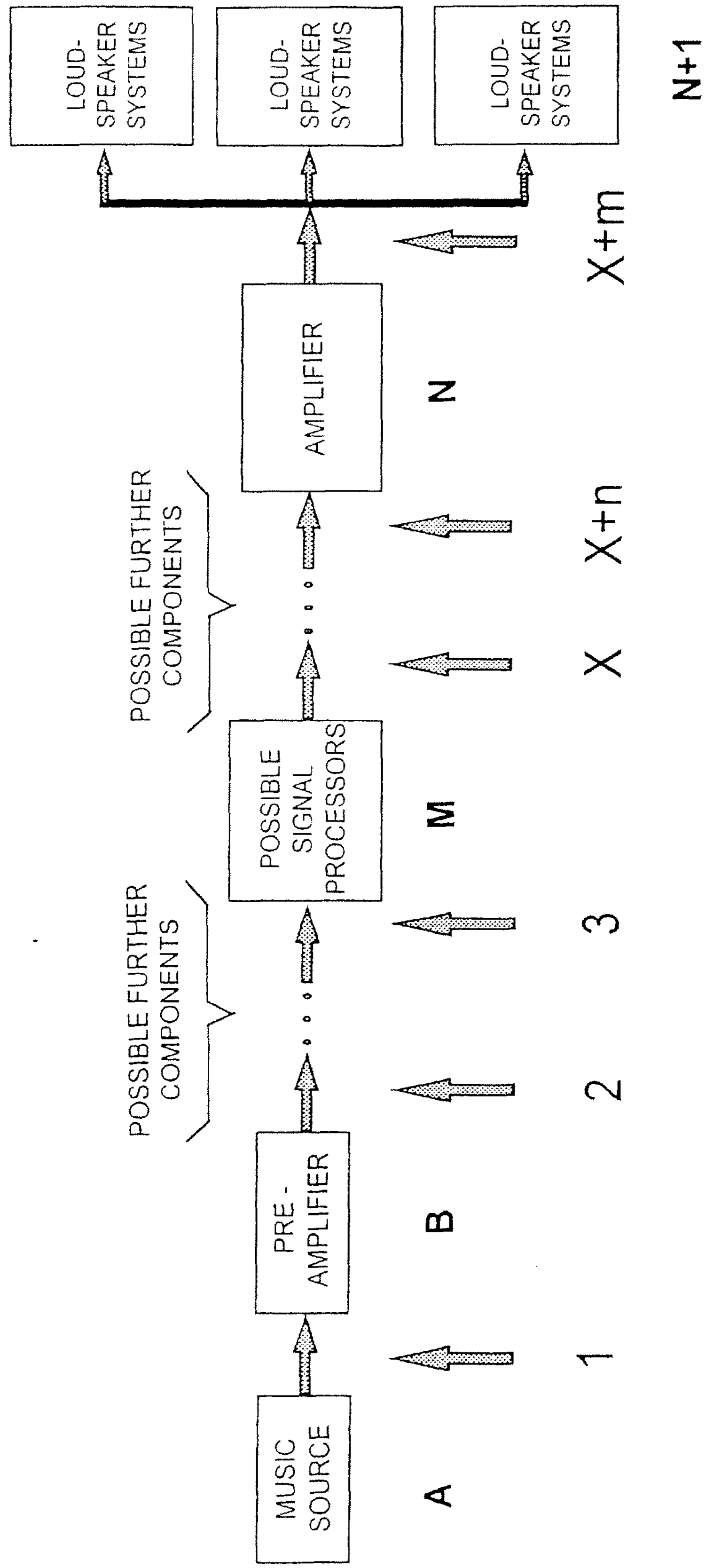

LOW TONES
SECTION
LOUDSPEAKER
MEDIUM / LOW
TONES SECTION
LOUDSPEAKER
MEDIUM / HIGH
TONES SECTION
LOUDSPEAKER
HIGH TONES
SECTION
LOUDSPEAKER
DYNAMIC
REGULATOR
RD1
DYNAMIC
REGULATOR
RD2
DYNAMIC
REGULATOR
RDn+1
DYNAMIC
REGULATOR
RDn+2
LOW-PASS
FILTER
FPB
BAND PASS
FILTER
FPB1
BAND PASS
FILTER
FPBn
HIGH PASS
FILTER
FPA
INPUT
I3

FIG. 5



MUSIC SOURCE
A
1
PRE - AMPLIFIER
B
2
POSSIBLE FURTHER COMPONENTS
3
POSSIBLE SIGNAL PROCESSORS
M
X
POSSIBLE FURTHER COMPONENTS
X+n
AMPLIFIER
N
X+m
LOUD-SPEAKER SYSTEMS
LOUD-SPEAKER SYSTEMS
LOUD-SPEAKER SYSTEMS
N+1

FIG. 6